Commodore Sound Studio User's Guide

SFX
COMPUTER SOFTWARE

commodore

INTRODUCTION

Welcome to the Commodore Sound Studio. This software package consists of two separate programs:

1. A Synthesizer Program that lets you create your own sounds using the Commodore's internal sound generators.
2. Music Programming software that enables you to produce your own musical compositions utilising both sounds derived from the synthesizer section, and from additional external musical instruments.

You thus have, at your fingertips, a comprehensive music production facility that offers most of the facilities of a multi-track electronic music recording studio.

CONNECTIONS

Before attempting to use this software, check that you have followed the correct setting-up procedures, as outlined in the Commodore Owner's Manual. The sound output will be heard from the speaker of your television or monitor.

Diskette

Before switching on, check that your Disk Drive is correctly plugged in and ready for use. Now switch on your Commodore and TV or monitor. The screen will display the READY signal with a flashing cursor.

LOADING THE PROGRAM FROM DISK

1. Put the diskette into the Disk Drive.
2. Type LOAD "*", 8, 1 then press RETURN.
3. After a few moments you will be given a choice of:
   F1/Synthesizer.
   F3/Editor.
4. Press F1 or F3 as appropriate. The program will now load.

N.B. When loading the Synthesizer, there is a pause between the disk drive stopping and the synthesizer screen appearing.

If you have selected Editor turn to Page 16 of this user's guide.

OVERLAY KEYBOARD

If you have the Music Maker keyboard overlay, place it on your computer so that it fits and operates correctly.

If you are not using the Music Maker keyboard the QWERTY keys may be used as shown in the table on the back page.

HI-FI

The sound of your Commodore can be enhanced by connection to your home hi-fi. You may connect your Commodore to your hi-fi using a 3- or 5-pin DIN plug. Use a screened lead with the shield connected to Pin 2 and the inner connected to Pin 3. The connections to the hi-fi end will depend upon your amplifier. For detailed information, consult your Commodore User's Guide.

GENERAL PROGRAM COMMANDS

LOAD SOUND

Press F1 and respond to the question at the top of the screen, SOUND LIBRARY No. ? by entering the sound number between 1-60, from the SOUND LIBRARY. Press RETURN and wait for a few moments.

SOUND LIBRARY

To view the contents of the SOUND LIBRARY, press SHIFT V.

SAVE SOUND

Press F3 and respond to the question at the top of the screen SAVE SOUND NUMBER by entering a number between 1-60. Press RETURN. You are now required to name your sound, by the prompt SAVE SOUND (No.) NAME, using a maximum of 9 characters. Type in the name, then press RETURN.

LIBRARY SAVE/LOAD (DISK UTILITIES)

Press 'V'. You are now given four choices:

L Load Library.
S Save Library.
D Directory of Libraries.
X Exit (return to synthesizer).

If you are not sure of the name of the library, Press D. After a moment a list of library names will appear.

To LOAD a library, press **L**, then type the library name in response to the prompt. Press RETURN.

To SAVE a library, press **S**, then type the library name. Press RETURN.

CURSOR UP/DOWN

Move the cursor (purple arrow) around the screen.

CURSOR LEFT/RIGHT

Adjusts the selected control.

D Draws the ADSR envelope.
M Demonstrates the sound.
< Octave down.
> Octave up.
X Exit to Sound Studio Editor.
H Help. Lists commands.
J Select joystick control.
F7 Press to select filter page.

SOUND STUDIO SYNTHESIZER

We regularly hear the term in connection with today's popular music, but what exactly is a synthesizer? Quite simply, a synthesizer enables you to create sounds, both musical and 'otherwise', by specifying a certain set of parameters. At any instant in time a sound has three distinct properties:

PITCH A sound could have a low pitch (e.g. a tuba or double bass), or a high pitch (e.g. a piccolo or penny whistle), or the pitch could fall anywhere in between.

TIMBRE The tone of a sound – it could be smooth and pure like that of a flute, or much brighter and tinkly like a harpsichord.

LOUDNESS Sounds can be at any volume level or loudness, from very quiet, to painfully loud.

These three fundamental elements determine the nature of what we hear, so by accurately defining each of them we can create an incredibly wide range of different sounds. And then by controlling them by means of the keyboard, we can use the sounds as a musical source just like any other instrument.

The Sound Studio software package enables you to do just this with the minimum of fuss and with maximum ease. But to really grasp the subject of Synthesis, one needs to understand a little of what 'Sound' is all about.

Q. What is Sound?

A. Sound is that sensation we experience when movement or vibrations in the air are detected by our ears. Our ears convert these vibrations into minute electrical pulses that are transmitted via our nervous system to the brain. The air around us is made up of billion upon billion of microscopic particles. You cannot see them, because they are so small, but they are there. These particles make up the atmosphere of our planet. And it is these air particles that transmit the sound from its source to our ear.

How? By moving backwards and forwards so as to form denser and less dense variations in the air. Take as an example a loudspeaker from a hi-fi system. Remove the front of the speaker cabinet, put on a heavy rock record (if you possess such a thing), and look closely at the cone of the large speaker. You'll see that it is moving backwards and

forwards very quickly, and usually you'll notice that it responds most dramatically every time the bass drum of the piece of music you are listening to sounds.

Consider what happens when we send a constant low pitched tone to the loudspeakers (figure 2). If the pitch is low enough, again we will see the cone vibrating. But what is happening with respect to the air particles?

(i) When there is no sound emanating from the loudspeaker, the air particles surrounding the unit are pretty well randomly distributed.
(ii) As the tone is introduced, the speaker shoots forward, compressing all the particles immediately surrounding the cone.
(iii) Having reached its maximum projection, the speaker then retreats back from its initial position, causing the air particles to become less densely packed (known as rarefaction).
(iv) The process continues to repeat itself until the situation shown in (v) exists.
(v) Here we have a train of compressions and expansions emanating from the loudspeaker, so if we were to consider an imaginary line drawn along the axis AA of the loudspeaker, at a given instant in time we could see how the speaker has caused the density of the air particles to vary.

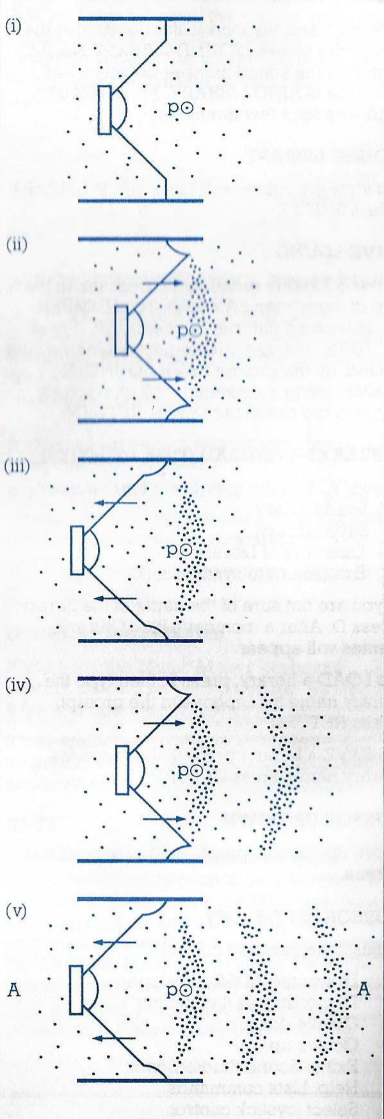
The ear detects changes in air pressure and it is these changes we perceive as sound. Our ears act like the loudspeaker did, but in reverse. The changes in the air pressure cause a membrane within the inner ear to move back and forth, causing a nervous signal to be sent to the brain. The above example of the loudspeaker is somewhat simplified; in reality the changes in air density caused by our rock music recording would be far more complex (rather like that shown in figure (3) where the more irregular shape is due to other instruments that would undoubtedly be incorporated into the rock music recording).

It should be noted that it is not the particles that move from the source to the listener. It is the variation in the air particle density that changes. This is illustrated by checking the movements of point **P** (which represent a specific air particle) in figure 2.

*It is the air particles that transmit sounds from source to receiver.

You are probably familiar with the schoolday experiment that proves the above statement.

Fig. 2



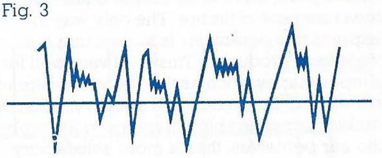


Fig. 3

A ringing alarm clock is placed in a bell jar, then all the air is pumped out leaving a virtual vacuum. As the air is removed, the sound becomes weaker and weaker until it is virtually inaudible. As soon as the air is restored to the jar, it is possible to hear the ringing again. Thus proving that it is the air particles that transmit the sonic information.

The Three Elements of Sound.

When we hear a sound, it can be defined at any one instant by considering three different parameters: the pitch, the timbre (from the French word for tone colour), and loudness (or amplitude). Unfortunately, it's not quite that simple, as you might be listening to an orchestra, where the sound emanates from many sources and combines. Here it would be necessary to make a composite analysis. It should also be stressed that these three parameters continually change throughout a note. Consider the following (figure 4). Here we are examining a period of, say, ten seconds. After one second a single note on a piano is played, and the changes in pitch, timbre and amplitude are detailed.

PITCH As you can see the pitch remains pretty well constant.

TIMBRE The tone starts off by being fairly bright, caused by the hammers of the piano striking the strings, and becomes more mellow as the note dies away.

LOUDNESS/AMPLITUDE The amplitude is nil before the note is played, it then shoots up to a maximum value as the note is played, and gradually decreases as the note fades away.

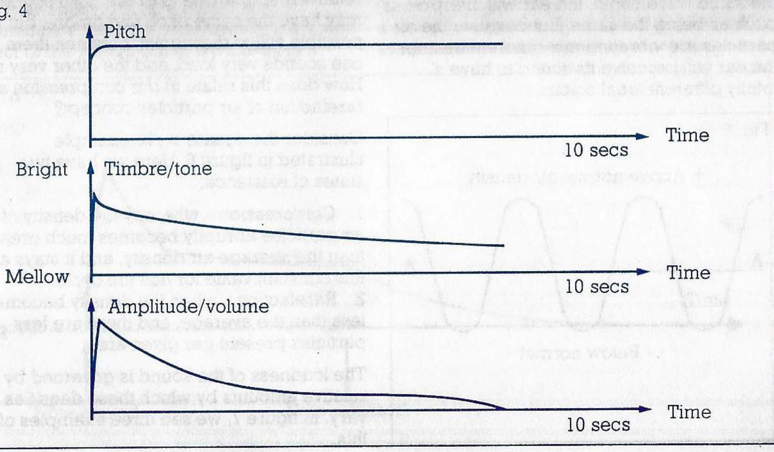
So if we can specify the pitch, timbre and amplitude of a sound at every given instant in time, we can create virtually any sound. This appears to be a long and laborious process, and so it would be if we had to break the sound down into fractions of a second and specify each parameter. But as we shall find, the design of most synthesizers is such that we are concerned with shapes, and these can be used to provide the changes in the sound quickly and efficiently.

Let's now look more closely at these three elements …

PITCH

In figure 5, we have introduced the concept of pitch and the waveform. The graphical

Fig. 4



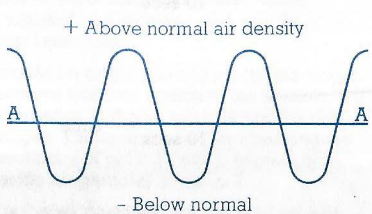
representation of how the density of the air particles varies with the distance from the loudspeaker, gives rise to the smoothly undulating shape depicted. As a tuning fork is oscillating back and forth, it constantly retraces the same curve. Waveforms are generally represented as variations of some parameter with respect to time. The horizontal axis is used to read off the time, and the vertical axis is the value of the parameter in question. Different shaped waveforms do not have the same tonal quality, but if they oscillate at the same frequencies then they will sound of similar pitch.

TIMBRE

Timbre is the quality of a sound that enables the listener to distinguish it from another of the same pitch. The timbre, or tone colour, of a note depends on the actual shape of the waveform produced. If we return to our example of the loudspeaker, we can see how the compressions and rarefactions of the air determine the shape of the waveform produced (figure 5). Look now at the way in which the air particles have lined up in figure 6.

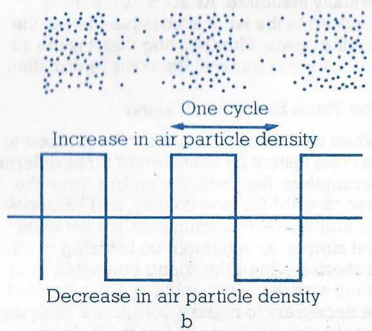
The source of the sound is such that the particles are compressed to a certain density for a set period of time, before being rarefied for an equal period. This is shown graphically in figure 6 (b). This waveform is known as a square wave. The wave still travels at the same speed (the speed of sound), so if it is of the same wavelength, the ear will interpret its pitch as being the same. But because the air particles are vibrating in a different manner the ear will perceive its sound to have a totally different tonal colour.

Fig. 5



Unlike pitch, there is no simple quantitive measurement of timbre. The only way to express this parameter is to describe the waveform produced. This is all very well for simple shapes, such as the two we've already mentioned, but since just a small variation can make a considerable difference to the timbre the ear perceives, then a more satisfactory method of describing this parameter is necessary. A waveshape can be defined by means of a mathematical equation, but to most of us, this is more an academic exercise than of any practical use.

Fig. 6



LOUDNESS/AMPLITUDE

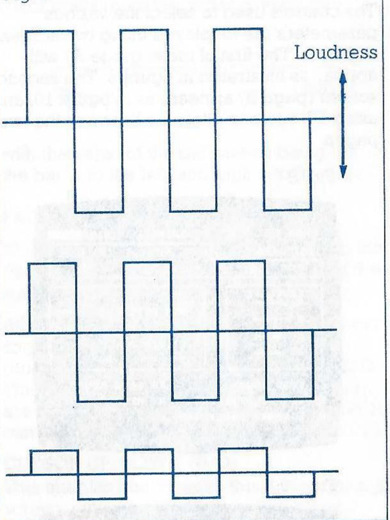
The concept of loudness is, on the surface, a relatively simple one to grasp. Two sounds may have the same pitch and timbre, but it is possible still to distinguish between them if one sounds very loud, and the other very soft. How does this relate to our compression and rarefaction of air particles concept?

Consider the square wave example illustrated in figure 6. Here we have two states of existence:

1. Compression – whereby the density of the air particles abruptly becomes much greater than the average air density, and it stays at this constant value for half the cycle.
2. Rarefaction – when the density becomes less than the average, and there are less particles present per given area.

The loudness of the sound is governed by the relative amounts by which these densities vary. In figure 7, we see three examples of this.

Fig. 7



The wavelength and waveshapes remain the same, just the variations in air densities are different, and the greater the difference, the louder we perceive the sound. The greater the peaks and troughs, the greater the amplitude or loudness.

When listening to music, loudness isn't as simple as loud and soft. There's much more to it. When considering a sound, the dynamics (or changes in loudness) are a vital aspect of the way in which we interpret it.

SOUND CHANGES WITH TIME

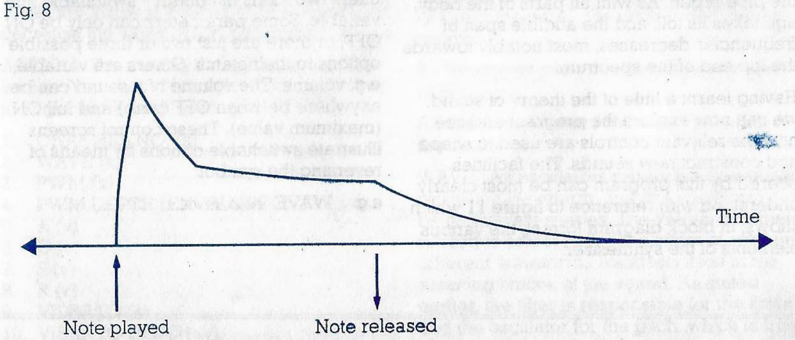
If you listen to the sound produced by almost any musical instrument, you will be aware that these three parameters vary during the course of the note.

1. LOUDNESS AND TIME All sounds have what is known as a contour, or envelope, i.e. they have a starting point from which their loudness goes from zero to a particular level. This level will then probably change during the course of the sound, and there will eventually come a point when the loudness decreases and the sound ends. So the sound produced by playing a note on a piano would have an envelope that started fairly abruptly as the hammer hit the strings, and which would die away gradually as long as the note was held, and would then fall away sharply as the key was released and the dampers deadened the strings. The contour, or envelope, shapes the loudness of the note (figure 8).

2. TIMBRE AND TIME The tone of most sounds changes during the course of the note. This is quite a complex phenomenon to illustrate, but we'll continue with the piano as our example. You can detect that the timbre of a note is much brighter when the key is struck; as the note dies away, the higher frequency elements of the sound tend to diminish even quicker, and the tone of the note becomes increasingly mellow.

3. PITCH AND TIME With most musical instruments the pitch remains relatively constant for the duration of the note. However, a small low frequency variation in pitch will produce the effect known as 'vibrato'.

Fig. 8



THE HUMAN EAR AND SOUND

The three elements of sound described above are nice, tidy concepts, all of which can be neatly evaluated mathematically. But then along we come with our ears, the receivers of the sound, and unfortunately, due to the limitations of the response of our ears' we have to make certain allowances when dealing with raw sound.

Our ears function rather like the speakers of a hi-fi system but in reverse. The compression and rarefactions of any sound that reaches our ears, cause a small membrane to vibrate back and forth, and this vibration is detected by a series of small hairs contained within our inner ear. These generate a minute signal which is transmitted via our nervous system to the brain, and hey-presto we hear noises. In essence it's very simple, but like a hi-fi system our ears have limitations as to the 'quality' and range of sounds to which they can respond.

PITCH AND THE HUMAN EAR

Our ears will respond only to frequencies within a certain range, known as the Audible Frequency Spectrum. Why? Because the actual mechanism of the ear can operate only in a certain range. But even the best microphones cannot detect sounds as efficiently as the human ear.

Generally speaking, the frequencies a healthy young adult can detect range from between 18-25 Hz up to around 20,000 Hz, which in musical terms is about a ten octave span – wider than virtually all acoustic musical instruments, with the possible exception of the pipe organ. As with all parts of the body, age takes its toll, and the audible span of frequencies decreases, most notably towards the top end of the spectrum.

Having learnt a little of the theory of sound, we can now explore the program and see how the relevant controls are used to shape and construct new sounds. The facilities offered by this program can be most clearly understood with reference to figure 11 which shows, in block diagram format, the various elements of the synthesizer.

GENERAL INSTRUCTIONS

The controls used to select the various parameters are displayed using two screens, or pages. The first of these (page A) will appear as illustrated in figure 9. The second screen (page B) appears as in figure 10, and, using F7, you can alternate between the two pages.

Fig. 9

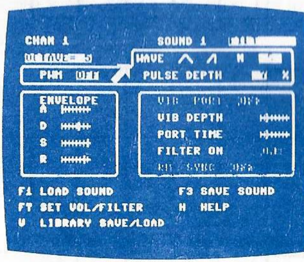
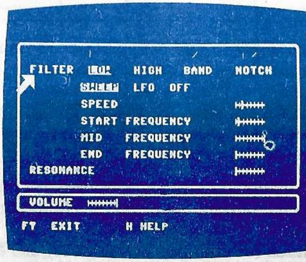


Fig. 10



Like any music synthesizer, this program offers two kinds of control – switchable and variable. Some parameters can only be ON or OFF, or there are just two or three possible options to their status. Others are variable, e.g. volume. The volume of a sound can be anywhere between OFF (zero) and full ON (maximum value). These control screens illustrate switchable options by means of reversing the symbol.

e.g. WAVE ∧ ⩘ N ⊓

Variables are depicted as if they were a horizontally mounted slider control thus:

———————1———————

min max

with the value of the parameter being low if the bar is to the left, and high if to the right.

PAGE A

This screen is designed to let you set up the Pitch, Initial Timbre and the Envelope of the sound.

ACTION. Press the keys, and you will hear a sound. As you can see, there is a large (purple) arrow pointing to the word WAVE. This arrow is indicating the parameter you are currently looking at, and whose value you can change.

CURSOR UP/DOWN (U/D)

This enables you to move this arrow down the screen, selecting the various individual parameters then cycling back to the first.

SHIFT + CURSOR UP/DOWN (Sh+U/D)

This moves the arrow back up the screen, cycling round the parameters the other way.

CURSOR LEFT/RIGHT (L/R)

When the arrow is a variable parameter, this control will increase the value of that parameter, and the 'slider knob' will move to the right. If a switchable parameter has been selected, this function key will cause the individual options to be selected (set).

SHIFT + CURSOR LEFT/RIGHT (Sh+L/R)

When the arrow is a variable, this control will decrease the value of that parameter.

ACTION Try moving the arrow around the screen. You should be able to get the arrow to alight on all the following:

1. CHANNEL NO. (s)
2. WAVE (s)
3. PWM (s)
4. PWM DEPTH (ENVELOPE) (v)
5. A (v)
6. D (v)
7. S (v)
8. R (v)
9. VIBRATO (s)
10. VIBRATO DEPTH (v)
11. PORTAMENTO SPEED (v)
12. FILTER ON (s)
13. RING MODULATION/SYNC (s)

(s) = switchable
(v) = variable

ACTION Utilising the LEFT/RIGHT CURSOR try altering both switchable and variable parameters - note the response on the screen.

JOYSTICK CONTROL

A joystick can be used to move the cursor round the screen and adjust the controls. Plug a joystick into port 2 and press J on the keyboard. Make sure 'Joystick On' appears on screen. Joystick Up/Down controls the cursor (purple arrow) while joystick L/R adjusts the controls.

Let's now look at the purpose of these 13 options with reference to figure 11. (This shows the basic block diagram of this page).

THE OSCILLATOR

The oscillator is the vibrating medium that determines the pitch of the note, but this pitch is governed by several factors:

1. The note played on the keyboard. This controls the oscillator over a two octave range.
2. The > and < buttons which are used to raise or lower the pitch in octave steps over a six octave range (displayed in the green OCTAVE = window of PAGE A).

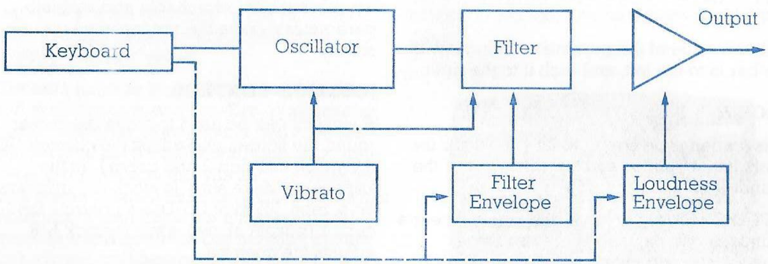
ACTION Play a note, then release it. Press > once. Play the note again - the pitch should rise an octave. Repeat the process until the note fails to rise any higher. Now press < and play the note - the pitch should fall an octave.

3. Vibrato or pitch modulation (dealt with later).

All these parameters will affect the pitch of the oscillator.

WAVE All oscillators produce a constantly repeating waveform, and this waveform can be of different shapes. The WAVE parameter is used to select them. The effect of selecting different waveforms manifests itself in the resulting timbre of the sound. As stated earlier, the filter is responsible for the timbre and the oscillator for the pitch, which is true.

Fig. 11 **BLOCK DIAGRAM OF SYNTHESIZER**



However, the waveshape produced by the oscillator also plays a part in determining the overall timbre of the sound, and it is logical for this control to be included in the oscillator section.

The synthesizer can produce four different waveforms:

(∧) Triangle
(⊿) Sawtooth (sometimes called Ramp)
(N) Noise
(⊓) Square/Pulse

ACTION Set arrow to WAVE and select the individual waveforms in turn using the LEFT/RIGHT CURSOR key. Try out each waveform using the keyboard. You should notice the following characteristics of the waveforms:

1. TRIANGLE This is a very smooth, mellow-sounding waveform. It is useful when creating flute-like sounds.
2. SAWTOOTH/RAMP This waveform has a much richer quality and a more brassy air to it.
3. NOISE This is an unpitched sound made up from a random mass of frequencies throughout the audio spectrum. It is useful in creating wind and sea sounds.
4. SQUARE/PULSE The Square/Pulse waveform is a particularly useful and interesting waveform requiring further discussion. This waveform option is used in conjunction with the set of controls in the box directly below.

ACTION Select the SQUARE/PULSE WAVE, which, when activated, displays **S**, then move the arrow to PWM. Using CRSR L/R ensure that this parameter is OFF, then move the arrow to PULSE DEPTH. Change this variable using CSR L/R. You will notice that the readout alongside moves from 1% to 50%. By holding CRSR L/R get the display to read 50; stop and play a note. You will notice that the sound is smooth and hollow – rather like a clarinet. You are listening to a Square Wave – see figure 6.

Now, using SHIFT + CURSOR L/R, reduce the readout and play a note; the sound will become much thinner and more nasal. This is because the square wave is being transformed into an increasingly thin pulse wave.

So by specifying the pulse width we can determine the 'quality' of the sound we're hearing. One of the nice things about these waveforms is that they make a very pleasing sound as their pulse width changes. This change in width can be effected automatically using the PWM (Pulse Width Modulation) facility.

Quite simply, all that is happening is that a very slowly cycling waveform (one running at around one cycle per second) is being used to vary the pulse width - as the waveform rises so the pulse width narrows; as the waveform falls, the pulse width widens. Note that the pitch of the note remains constant - just the pulse width varies.

ACTION Select the square wave option, set PWM to ON, and increase the 'PULSE DEPTH' to maximum (50) using CRSR L/R. Play a note. You will now hear a much fuller and richer timbre. Move the arrow back to PWM and press CRSR L/R to turn the parameter OFF - you will instantly notice how the sound has a much flatter quality.

The pulse width modulation facility is extremely useful when creating powerful sounds, but should be used sparingly when utilising the multi-track facility. If all tracks sound strong and rich in harmonics then you lose subtlety in the composition. Having set up the pitch and shape of the basic waveform, it is necessary to consider the main modulation controls. These are housed in the box to the right of Page A and appear as in figure 9.

The first option:

VIB PORT OFF

is used to select either Vibrato, Portamento or no modulation at all.

VIBRATO

When a violinist plays a long note, you will often notice that he introduces a constantly repeating change in the pitch of the note. He does this by rocking the finger holding down the string back and forth. This effectively lengthens and shortens the string by a small amount, and this in turn raises and lowers the pitch a little. You will hear opera singers modulating their voices so that towards the end of the note, the pitch varies. This effect is known as vibrato - it's the symmetrical variation of pitch with time.

ACTION. Making sure that PWM is set to OFF (in order to make the effect more prominent), select VIB, then move the arrow to VIB DEPTH and move the pointer to the mid position. Play a note. Notice how the pitch wavers very quickly, moving up and down a small amount two or three times a second. By increasing or decreasing the depth you can make this effect more, or less, prominent. Vibrato is a very useful effect, which again should be used sparingly.

PORTAMENTO

This effect gives the synthesizer an advantage over all other types of keyboard instruments in that it allows the player to move in between notes. It is best explained by example:

ACTION. Select PORT. Move the arrow to PORT TIME and move the control to the central position. Play the top note of the keyboard then the bottom. Instead of hearing two distinct notes you will hear the pitch slide up to the top C then slide all the way down to the bottom note. Portamento is the sliding of the pitch between notes. Normally, when we press a series of notes, we get distinct pitches. With this facility, the synthesist can slide between notes in the same way as the trombonist, and other acoustic musicians can.

RM (RING MODULATION)

Ring Modulation takes two sounds and combines them so as to make two new ones - the new sounds have pitches that correspond to the sum and the difference of the original signals. So say we have two sounds of frequencies a and b. When these sounds have been ring modulated, the new sounds are of frequencies $a + b$ and $a - b$.

The synthesizer ring modulates the sound that you have created with another internal sound, to create a new pair of sounds.

SYNC (Synchronization)

The Synchronization effect again requires two sound sources to function. In this case, the frequency of one signal locks on to a harmonic of the other. The effect of Synchronization (and Ring Modulation) is best understood by practical experiment. The sound is created in a similar way to Ring Modulation.

N.B. Note that Ring Modulation and Sync can only be used in the multi-track recording section when the notes on tracks 2 and 3 are the same, as the effect is derived from two sounds.

FILTER ON

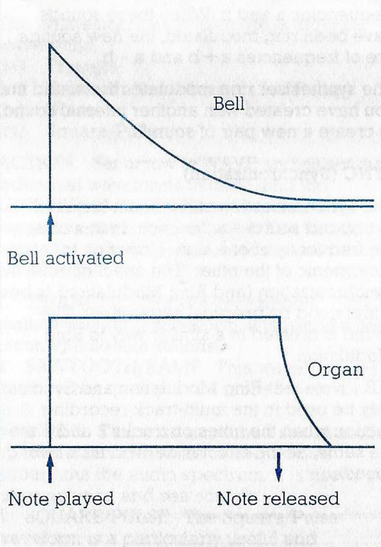
Also located in this control box is a parameter marked FILTER ON. When in the OFF position, the filter is bypassed, and no further tonal variation can be made to the sound. In the ON position the Filter section's controls, located on Page B, become active.

ENVELOPE

On this page are to be found the loudness envelope controls. An envelope is a control signal that is activated every time a key is played, and in this case it is used to shape the loudness of the note. Imagine the sound of any musical instrument – its loudness is constantly changing, if it weren't you would hear the note indefinitely.

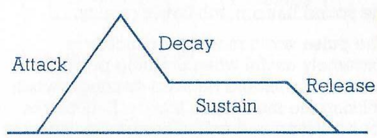
Figure 12 shows two envelopes 1. a bell and 2. an organ. What you are seeing is how the sound varies with time. With the bell, there is silence before it is struck, then the amplitude virtually instantaneously rises to a maximum, then the sound gradually dies away. While with the organ, sound rises to a maximum volume the instant a note is played, and remains at that level until the key is released,

Fig. 12



whence it falls back to nothing. The ENVELOPE section enables us to create these types of waveform and use them to control the loudness of the tone produced by the oscillator. The trigger for the waveform is the playing of the note on the keyboard. Four variable controls enable us to set up virtually any envelope shape we might need. This is known as an ADSR envelope, and we can explain it with reference to figure 13.

Fig. 13



A. The Attack time. The period the envelope takes to reach its maximum level after a note has been played.

D. The Decay time. The period that it takes for the envelope to fall from its maximum point to…

S The Sustain level. The amplitude to which the envelope decays, but only if the note is still being held. Once at this level the envelope will remain at this point until the key is released.

R. The Release time. The period it takes for the envelope to fall away from the sustain level, or from whatever level the envelope was at when the key was released, back to its original zero value.

VISUAL DISPLAY OF ENVELOPE

In order to be able to get a better idea of how changes in the various envelope parameters affect the final shape, a graphic envelope display facility has been incorporated into the Sound Studio program. By pressing **D** you will see a graphic representation of the envelope along the bottom of the page.

THE FILTER

We now move on to consider an extremely important aspect of the Sound Studio software – the filter. But before doing so, select Page A, as shown in figure 9, ensuring that the FILTER ON control is set to ON.

ACTION. PRESS F7. We have now turned to Page B – Page A can still be accessed by pressing F7 again. From our block diagram figure 11, we can see that the signal from the oscillator is fed to the filter. This program filter is in effect a form of tone control, just like the treble and bass knobs on your hi-fi system, but here we have considerably more control to determine the intricate character of the sound being created. In fact the Sound Studio's filter section is more elaborate than that found on most commercially available dedicated synthesizers.

The Sound Studio filter does the same job as any other type of filter – it removes part of the 'material' being passed through it. A tea strainer is an example of one kind of filter, it holds back the tea leaves while letting the liquid through. Similarly, a fisherman's trawl net will let fish up to a certain size slip through, but catch the larger ones. The types of filter that are used in this and other electronic music synthesizers remove certain frequencies from the signal fed through them, and it is the filtrate, or remaining signal, that is left as the desired sound. Along the top of Page B are four (red) symbols labelled underneath:

LOW HIGH BAND NOTCH

The Sound Studio enables you to select any of these four main types of filter.

1. The LOW pass filter: this will remove all frequencies above a certain frequency, hence it lets low frequencies pass.
2. The HIGH pass filter: this will remove those frequencies below a certain point.
3. The BAND pass filter: this will let through only those parts of the signal that are of a certain frequency.
4. The NOTCH filter: this will remove from the signal frequencies of a certain value.

N.B. Press **M** for a demonstration of the selected effect.

Fig. 14

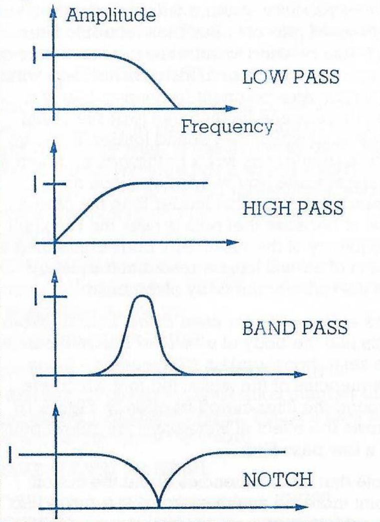


Figure 14 illustrates these filter configurations diagrammatically, and also shows the corresponding symbol depicted on the screen. Imagine that you were feeding the filter with a signal that consisted of every frequency, all with the same amplitude (like a white noise source). The filtered signal would thus appear as in the four diagrams with the vertical axis representing the amplitude of the frequencies, and the horizontal axis measuring off the frequency.

Consider the low pass filter. We can see that the graph is flat up to a certain point, then it starts to trail off until the amplitude, of the higher frequencies becomes zero. The point, i.e. the frequency, at which the amplitude starts to become attenuated is known as the cut-off frequency.

ACTION. Select the low frequency filter option. Ensure that the SWEEP option is selected. Move the arrow to START FREQUENCY.

By altering this parameter, you are in effect changing the filter cut-off frequency and you will hear sounds with greater and lesser amounts of high frequencies. Experiment using the other types of filter i.e. high pass, band pass and notch.

RESONANCE

When you filter a sound you remove an unwanted part of it. But, the electronic filter can also be used to increase the resonance of a sound. Most musical instruments have what is known as a resonant frequency, that is to say there is one (sometimes two) particular pitches at which they sound louder. If you go into a small room, say a bathroom, and hum a musical scale, you will find that one note sounds considerably louder than the others – that is because that note is near the resonant frequency of the room. Any body containing a mass of air will have a resonant frequency – be it a room or the body of the cello.

The electronic filter used in the 'Sound Studio' acts like the body of a cello or guitar, it can be set to boost certain frequencies – those frequencies of the signal fed to it which are around the filter cut-off frequency. Figure 15 shows the effect of increasing the cut-off point of a low pass filter.

Note that the frequencies about the cut-off point increase as the resonance is turned up.

FILTER MODULATION

When we were discussing the fundamentals of synthesis, we stated that the timbre of a sound usually varies over its duration. So in order to achieve this with our Sound Studio software, it is necessary to be able to modulate the filter cut-off frequency.

The line:

SWEEP LFO OFF

enables us to select this facility.

LFO

When selected, the slowly cycling oscillator, as used for Pulse Width Modulation, is routed so as to vary the cut-off frequency of the filter. This results in the continual change of the cut-off frequency. This effect is useful for more 'spacey' sounds as well as a more subtle form of vibrato.

ACTION. Ensure that the filter is ON (Page A) then select LFO (Page B). Play a note and you will hear the effect.

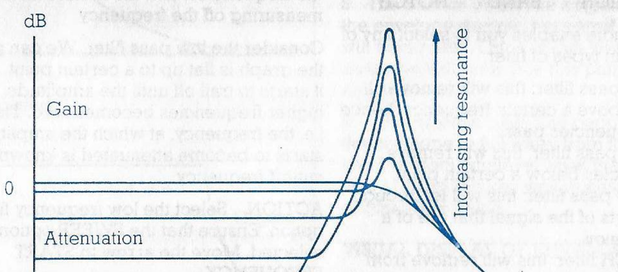
SWEEP

Enables us to feed an envelope to the filter, thus we can automatically change the timbre of the sound over the duration of the note. In this case, the envelope is a little different from the one used to shape the loudness of the sound. The envelope's shape is set using the following variable controls.

STEP SIZE
START FREQUENCY
MID FREQUENCY
END FREQUENCY

STEP SIZE. This determines the duration of the envelope used to modulate the filter cut-off frequency. When set to a maximum value,

Fig. 15



the cut-off frequency will take a longer time to trace out the patch set by the filter envelope.

The filter envelope is defined using three points of reference:

START FREQUENCY - is the initial setting of the cut-off point when the key is played.

MID FREQUENCY - is the frequency to which the envelope moves at a rate determined by the STEP SIZE.
END FREQUENCY - having reached the MID FREQUENCY the filter's cut-off point moves on to this setting also at a rate determined by the STEP SIZE.

ACTION. Select Ramp wave, turn the FILTER ON, and set up the following envelope (Page A).

A 1————————————————
D ——————————————1————
S ————————————————1
R ————————————————1

then set the filter controls as below.

FILTER LOW

SWEEP

STEP SIZE 1————————————
START FREQUENCY 1————————————
MID FREQUENCY ————————————1
END FREQUENCY ————————————

RESONANCE ——————————1————

The RESONANCE is set high in order to accentuate the effect. Play and hold a note. You will hear the timbre slowly become brighter until it reaches a maximum, then become more mellow until it reaches the END FREQUENCY setting. REPEAT the above, but with the STEP SIZE set fully to the right. Note that the duration of the envelope is greatly reduced.

VOLUME

The final control on this page is the master volume. This simply sets the overall level of the sound. As you can appreciate, some sounds will seem quieter than others by virtue of the amount of filtering they've received. So in order to enable a balance to be set up between sounds, it is initially worth setting this control to the mid position, then varying it to bring it into line with the other sound being used.

STORING A SOUND IN MEMORY

Having mastered all the voice production controls of the Sound Studio system, you can now proceed to create some sounds of your own - so EXPERIMENT!

You will find that you will soon come up with a certain sound that you think sounds great, and which you want to use at a later date. The Sound Studio software has a sound library which has a host of sounds already in it, and which also has room to store your own sounds.

ACTION. Having set up a particularly good sound, select Page A. PRESS F3. At the top of the screen you will see:

SAVE SOUND NUMBER

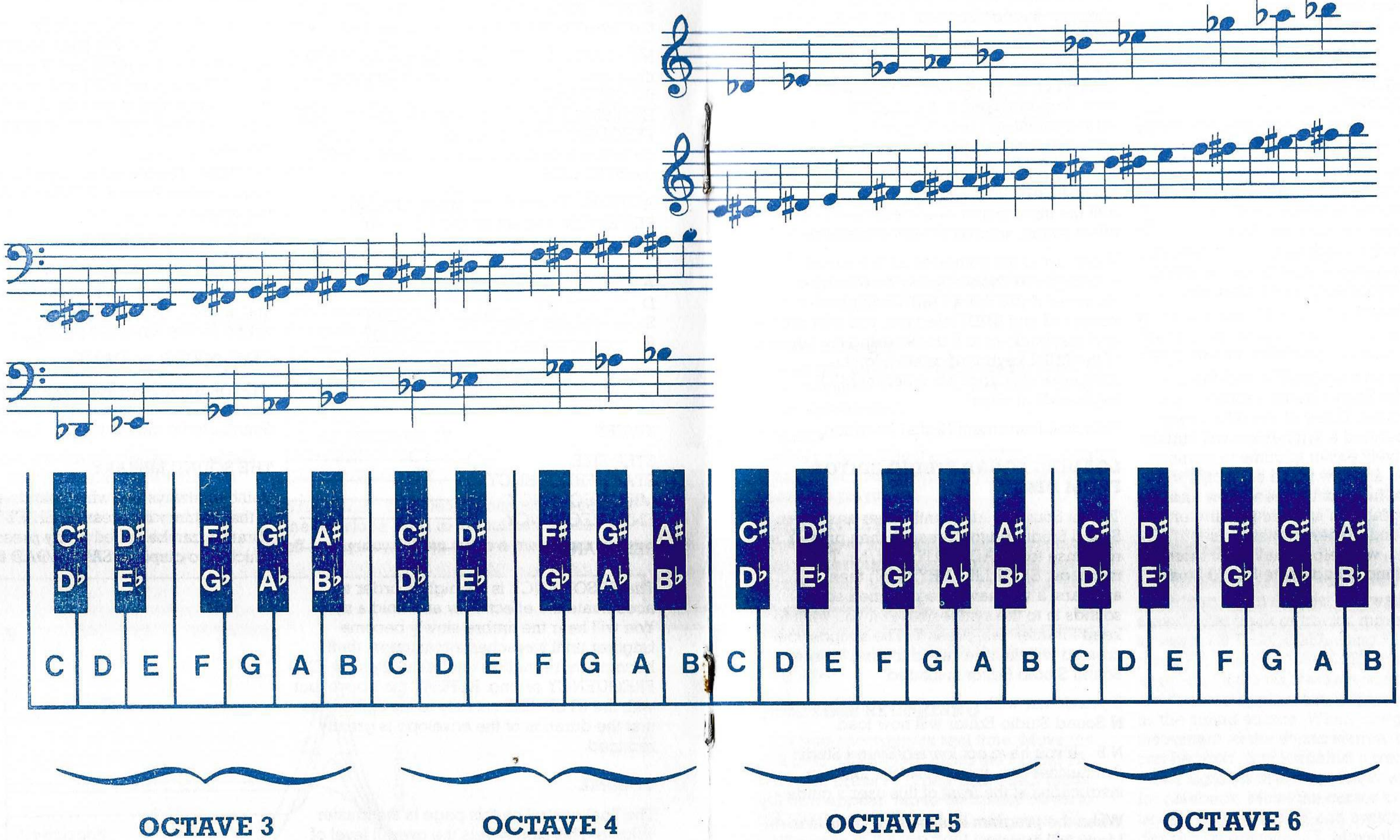
TYPE in a number, (n), between 1 and 60. (This will be the memory location number of that sound.) PRESS RETURN. You will then be asked (at the top of the page):

SAVE SOUND (n) NAME

TYPE in a name for your sound. PRESS RETURN. Your sound is now stored in the Sound Studio memory.

THE SOUND LIBRARY

You can always see what sounds are stored in the memory by pressing SHIFT **V**. Sound libraries can be called up by pressing **V**, which also displays SAVE/LOAD functions.


C♯ D♭ D♯ E♭ F♯ G♭ G♯ A♭ A♯ B♭
C D E F G A B
C♯ D♭ D♯ E♭ F♯ G♭ G♯ A♭ A♯ B♭
C D E F G A B
C♯ D♭ D♯ E♭ F♯ G♭ G♯ A♭ A♯ B♭
C D E F G A B
C♯ D♭ D♯ E♭ F♯ G♭ G♯ A♭ A♯ B♭
C D E F G A B
OCTAVE 3
OCTAVE 4
OCTAVE 5
OCTAVE 6

THE KEYBOARD

When you play a note on a musical instrument such as a piano, what is happening?

1. You are selecting one particular note on the keyboard, thus specifying the basic pitch of the sound.
2. When you actually press down the note you are telling the instrument 'I want the sound to begin now'.
3. When you release the note, you are saying '... I want the note to finish now'.

The role of the keyboard therefore is to say when a note starts and finishes, and what that note is. The former is known as 'Gate' information, and the latter as 'Pitch' information. In the block diagram (figure 11) the pitch information is sent to the oscillator, and the gate information to the filter and loudness envelope.

SUMMARY

You should by now be familiar with the workings of the Sound Studio's sound creation functions. Those of you who have previously operated a synthesizer will find the controls relatively easier to come to terms with, but if you are new to the subject, it will take a while to fully comprehend how and why the changing of a specific parameter affects the sound. Experimentation is important! This will help to make you more aware of the functioning of the Sound Studio synthesizer.

WHAT IS SOUND STUDIO EDITOR?

Sound Studio Editor is a powerful program which enables you to create a complete multi-track arrangement with different sounds on each channel, volume, sound and tempo changes during playback, and repeats of any section.

A piece of music can be input in 'real time' (played on a keyboard), one channel at a time, then replayed as a complete arrangement, or input in 'step time' note by note, like a word processor.

Another powerful facility is the ability to input music in 'real time' then go to 'step time', and edit the input, either to correct mistakes or to effect sound, volume or tempo changes.

When using the computer as the sound source, three channels may be recorded. However if you use a *MIDI equipped keyboard and MIDI interface, you may record and playback up to 6 tracks using the sounds of the MIDI keyboard or even let the computer play up to six different MIDI keyboards at once!

*Musical Instrument Digital Interface.

LOADING SOUND STUDIO EDITOR FROM DISK

To exit Sound Studio Synthesizer and enter Sound Studio Editor, press **X**, then press **Y** in response to: LOAD EDITOR (Y/N). The message: SAVE LIBRARY (Y/N) then appears. If you have programmed some sounds in to the synthesizer that you wish to keep for later use, press **Y**. The sounds you created will then be saved to disk, before Sound Studio Editor is loaded.

If you do not wish to save any sounds, press **N**.Sound Studio Editor will now load.

N.B. If you have not loaded Sound Studio Synthesizer first, then follow the loading instructions at the front of this user's guide.

When the program has loaded, the Editor Menu will appear.

EDITOR MENU

PLAYBACK (P.B.)

Play back any tracks which are 'switched on' (see P.B. track select).

REAL TIME
Enables you to record music as you play it, using the computer keyboard or the Music Maker overlay keyboard.

STEP TIME
Think of it as a musical word processor. You input notes and rests with their note lengths and pitch, plus sound, volume and tempo changes, one step at a time, then go back and change it if you wish. You can also edit a recording you have made in real time.

P.B. TRACK SELECT
Switch on or off the tracks that you wish to play back.

DELETE ALL MUSIC
Clears all tracks of all data.

MIDI
Musical Instrument Digital Interface. Set up options for output to or input from, a MIDI keyboard.

SOUND LIBRARY
Call up the library of 60 sounds supplied, or your own library of sounds from the Sound Studio Synthesizer.

TUNING
Enables you to tune the pitch of the computer to any other instrument, or *vice versa*, for ensemble playing.

DISK UTILITIES
Allows you to save your music on to disk and load in previously saved music. You can also format a disk and delete or rename a file.

LOAD SYNTHESIZER
Use this option to exit Sound Studio Editor and enter the Sound Studio Synthesizer program.

REAL TIME RECORDING

Let's start recording in real time. Move the cursor to REAL TIME using the F1 and F3 keys and press F7. The REAL TIME menu will now appear. Move the cursor down to RECORD, using F1 and F3, and press F7. The metronome in the top right hand corner of your screen will now start. (The metronome must be ON for real time recording).
Recording commences as soon as you start playing a tune on the keyboard, (one note at a time). It is recorded exactly as you play. Press F7 to stop recording.

To listen to your recording, move the cursor to PLAYBACK using F1 and F3 and press F7. If you are satisfied, you can now record another track to accompany the first one.

To record a further track, move the cursor to TRACK SEL, using F1 and F3 and press F7. Now select a different track (signified by the numbers 1-6) using F1 and F3 and press F7. Repeat the recording procedure as above to record a second track. The metronome will give you a count of one bar before recording starts. If your first track is switched on for playback, (see PB track sel), you will hear it playing as you record your second and subsequent tracks. Up to three tracks can be recorded in this way (six tracks when using MIDI).

SEL TIME SIG – time signature is set to 4/4 time when you first enter the program, but is easily changed by moving the cursor to 'sel time sig' using F1 and F3 and pressing F7. The time signature menu will now appear, giving you the following options:

3/4
4/4
5/4
6/4

Move the cursor to the desired time signature using F1 and F3 and press F7, the marker will move to your selected time signature. Move the cursor to exit and press F7 to return to the REAL TIME menu.

PB TRACK SELECT, if you have recorded more than one track, but you want to listen to a particular track or tracks, move the cursor to select PB TRACK SEL using F1 and F3 and press F7. The PB TRACK SEL menu now appears. Of the six tracks available, tracks one to three apply when using the computer as the sound source. When using a MIDI instrument as the sound source, tracks 1-6 can be used. A tick against a track on the menu signifies that the track is 'switched on' for playback. Move the cursor to the desired track using F1 and F3 and switch on or off as desired by pressing F7. Move the cursor to EXIT using F1 and F3 and press F7 to return to the REAL TIME menu.

TEMPO – to speed up or slow down the speed of playback of your music, move the cursor to TEMPO using F1 and F3 and press F7. To adjust, use F1 and F3. The number displayed between 44 and 208 corresponds

to normal metronome settings. PLEASE NOTE, when recording in 'real time' using a MIDI instrument, it may be necessary to set a fairly fast tempo if you experience difficulty recording short note lengths, due to differing MIDI standards on different instruments.

MIDI - to record in REAL TIME using a MIDI instrument, it is necessary to plug in a MIDI interface.

YOU MUST SWITCH OFF YOUR COMPUTER BEFORE PLUGGING A MIDI INTERFACE INTO THE CARTRIDGE PORT OF YOUR COMPUTER!

Reload the Sound Studio Editor program as described in the loading instructions. When loaded, the main menu appears. Move the cursor to MIDI using F1 and F3 and press F7. The MIDI menu now lists the following options:

MIDI - on or off.

OMNI - the music information of all six tracks is sent and received down one channel. All six tracks will therefore play the same sound on the MIDI keyboard.

POLY - selecting this option produces the POLY CHANNELS menu. This enables you to playback each track on a different MIDI instrument by assigning a different channel number to each track. You must match the channel number on the receiving instrument to the channel number assigned to the track you wish it to play (see your MIDI keyboard user's guide).

MONO - applies only to certain MIDI keyboards such as the CASIO CZ101.
Enables you to assign a different sound to each track on the same MIDI keyboard (see your MIDI keyboard user's guide).

P.B. TYPE 1 - to playback on MIDI keyboards such as the YAMAHA DX7 or SIEL MK900.

P.B. TYPE 2 - to playback on MIDI keyboards such as the CASIO CZ101.
RECORD TYPE 1 - to record (real time only) from MIDI keyboards such as the YAMAHA DX7 or CASIO CZ101.

RECORD TYPE 2 - to record from MIDI keyboards such as the SIEL MK900.

RECORD TYPE 3 - to record from MIDI keyboards such as the SIEL DK600.

Having set up your MIDI option return. to the Editor Menu by moving the cursor to exit using F1 and F3 then press F7. To record in 'real time' follow the instructions as before but use your MIDI keyboard to input your music.

DELETE TRACK - to delete the track currently selected, move the cursor to 'delete track' using F1 and F3 and press F7.

OCTAVE - to play your music at a higher or lower pitch, move the cursor to 'octave' using F1 and F3 and press F7. Now use F1 and F3 to select a higher or lower octave. Press F7.

STEP TIME RECORDING

Remove the Music Maker overlay if fitted. Move the cursor to 'step time' using F1 and F3 and press F7. The STEP TIME SEQUENCER page now appears. The large box on the left of the screen is the data window. The red bar across the middle is the position at which music is entered. Below this window is the box where the note data is input, together with a diagram showing the note values and the corresponding keys used for input. At the top right of the screen is a menu with the following options:

PLAY - play back selected tracks at any time.
TRACK SEL - used to select the current track for input.
P.B. TRACK SEL - choose which tracks you want to play back.
SOUND - choose whether or not to have notes sounding during input.
BLOCK DEL - use to delete large portions of data.
MIDI - switch MIDI on or off.

Below the step time menu is a reference list of commands used for music input as follows:

↑	=	♩³(triplet)
*	=	♩. (dotted note)
space bar	=	rest
plus (+)	=	# (sharp)
minus (-)	=	♭ (flat)
@	=	Clear # ♭ (clear sharp/flat)
shift T	=	Tie (tied note)

ENTERING MUSIC IN STEP TIME

Music input is easy. You must enter three pieces of information.

1. The name of the note e.g. C, A, F sharp etc. This is done by pressing the corresponding key on the QWERTY keyboard (not the music keyboard). Sharps and flats are entered by using the + and - keys respectively.
2. The octave in which the note is to sound e.g. 'C' in octave '5' is equivalent to middle C on a piano (see chart on page 14). To enter simply press the appropriate number on the keyboard, 1-9.
3. The note value. To enter, press the shift key and a number between 1 and 6 according to the chart below (and as shown on the screen):

1 = demi-semi-quaver (thirty-second note).
2 = semi-quaver (sixteenth note).
3 = quaver (eighth note).
4 = crotchet (quarter note).
5 = minim (half note).
6 = semibreve (whole note).

To enter a rest, press the space bar, then the note value equivalent to the length of rest you require.

If you make a mistake at this stage just re-enter the data correctly.

When the note data is assembled in the data box, press RETURN. The note data now appears in the red bar in the middle of the data window.

You will notice that the note data still appears in the data box. This is useful when you want to enter the same note data more than once - (just press 'return' again), or if you want to change only one or two parts of the data. This speeds up input tremendously.

DELETE NOTE DATA - To delete the last entered note data, press the delete key. To delete note data previously entered, scroll through the data using the 'cursor down' and 'cursor right' keys until the data you want to delete appears in the red bar across the data window. Now press the delete key.

To INSERT note data, scroll through the note sequence as above until you reach the point at which you want to insert new data, (this will be inserted AFTER the data which appears in the red bar). Then press RETURN.

To REPLACE note data, scroll through the note sequence as above until you reach the data you want to replace. Enter your new data into the data box, then hold the shift key and press RETURN.

In addition to note information you can also enter commands to effect tempo, volume and sound changes during playback or to repeat certain sections.

To effect a change in tempo, scroll through the note sequence to the point at which you wish the tempo to change. Press **T** followed by a number between 44 and 208 and press RETURN.

To effect volume changes during playback, scroll through the note sequence to the point at which you wish to change the volume. Press **V** followed by a number between 0 and 15, then press RETURN.

To effect sound changes during playback, scroll through the note sequence to the point at which you wish to change the sound. Press **S** followed by a number between 1 and 60 (see Sound Library), then press RETURN.

To repeat a section of music, scroll through the note sequence to the point at which you wish the repeat to start (repeat marker will be inserted immediately after the note data which appears in the red bar). Press **R** followed by RETURN. Now scroll through to the point at which you wish the repeat to end. Press **R** followed by RETURN. The message NO REPEAT NUMBER is now displayed at the bottom left of the screen. Enter a number corresponding with the number of times you wish the section to be repeated, then press RETURN.

If you wish to save time, instead of scrolling through a long sequence, press **L** followed by the line number you wish to go to, then press return. This is possible only up to line 999. The maximum number of steps that can be programmed is 3450. Beyond that the program's performance will be unpredictable.

LOADING AND SAVING MUSIC

Accompanying the Sound Studio program on your disk are two musical arrangements, 'Paganini's "Caprice"' and 'House of the Rising Sun' which can be loaded into the Sound Studio Editor, and played back or edited. You can also save your own musical arrangements (and sound library from the synthesizer program).

To load music from disk, select DISK UTILITIES from the EDITOR MENU using F1 and F3 then press F7. The DISK UTILITIES menu now appears. Move the cursor to LOAD using F1 and F3, then press F7. The menu titled TYPE OF FILE will now appear, with a choice of MUSIC or VOICE. Select MUSIC using F1 and F3, then press F7. After a few seconds, a directory of music on file will appear. Choose the one you wish to hear using F1 and F3, then press F7. A few seconds later the EDITOR MENU reappears.

To hear the music, select PLAYBACK (PB) using F1 and F3 then press F7.

To edit the music, select STEP TIME using F1 and F3, then press F7. Follow the instructions for ENTERING MUSIC IN STEP TIME.

To save music on to disk, select SAVE from the DISK UTILITIES menu. A menu titled TYPE OF FILE will appear from which you choose MUSIC using F1 and F3, then press F7. The SAVE FILE menu now appears. To give a file name to your music, move the cursor to the different letters of the name using F1 and F3 and enter each letter by pressing F7. To delete a letter, move the cursor to DEL and press F7. When the name is complete, move the cursor to END and press F7. The EDITOR MENU will reappear after a few seconds.

DISCLAIMER

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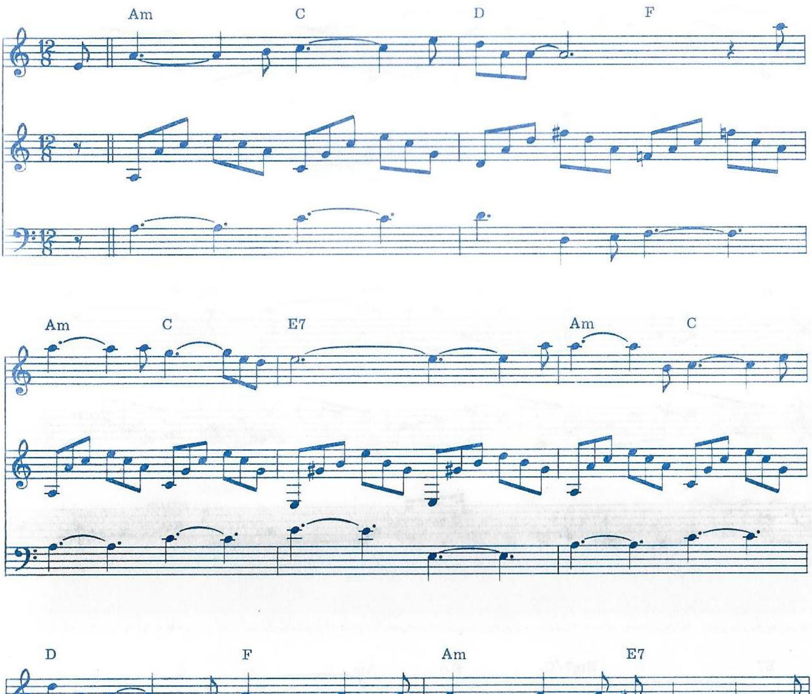
HOUSE OF THE RISING SUN
TRADITIONAL



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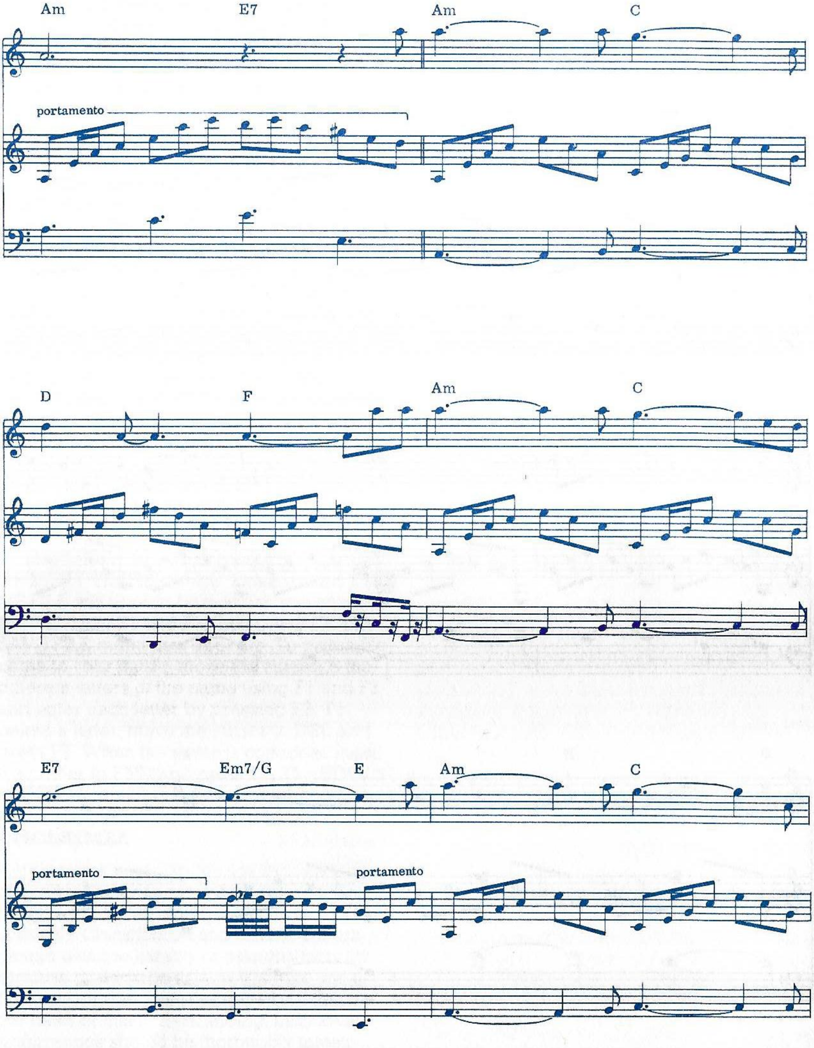
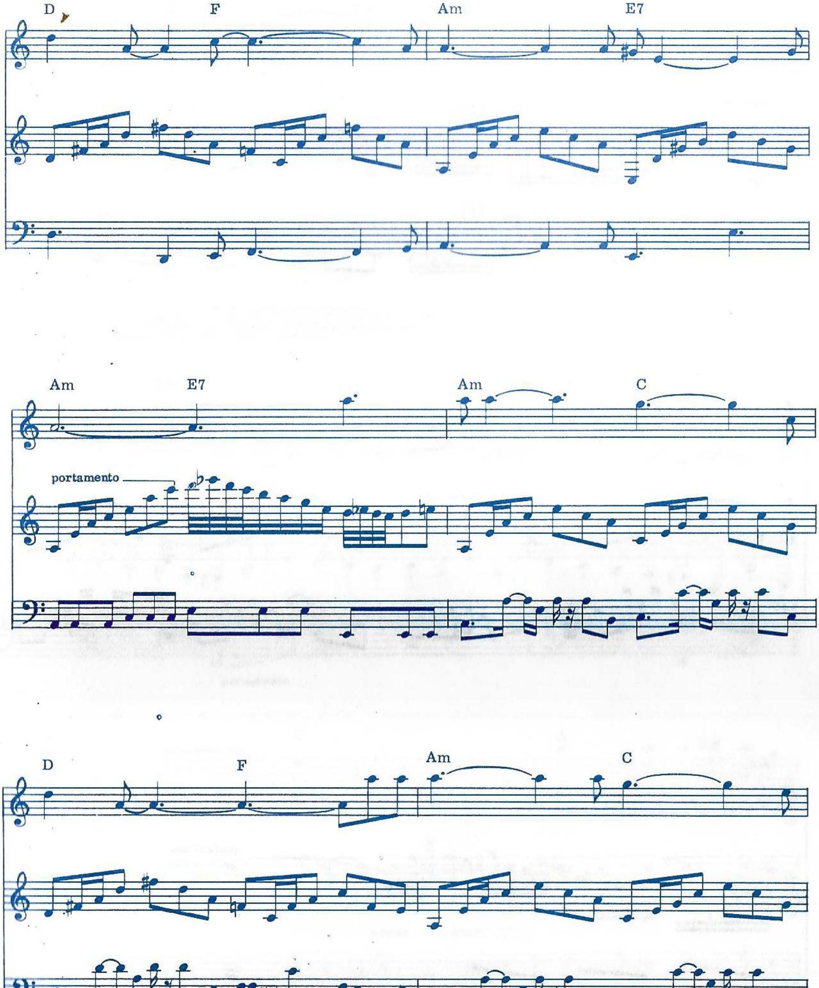

Am
E7
Am
C
portamento
D
F
Am
C
E7
Em7/G
E
Am
C
portamento
portamento


D
F
Am
E7
Am
E7
Am
C
portamento
D
F
Am
C


E7
Am
C
(♮)
(♮)
D
F
Am
E7
Am
portamento

CAPRICE
NICCOLO PAGANINI

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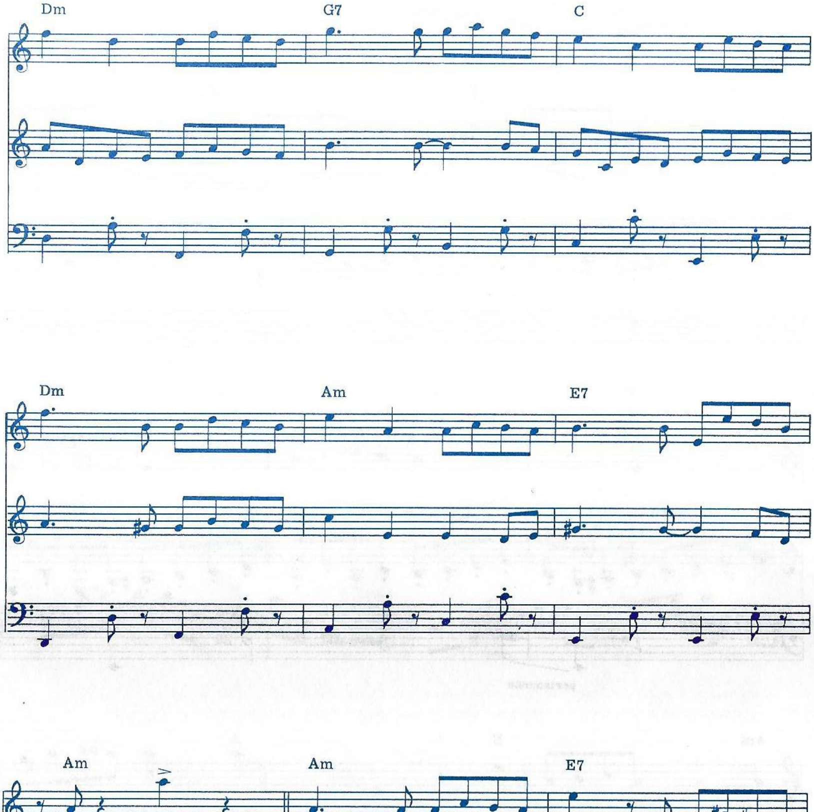
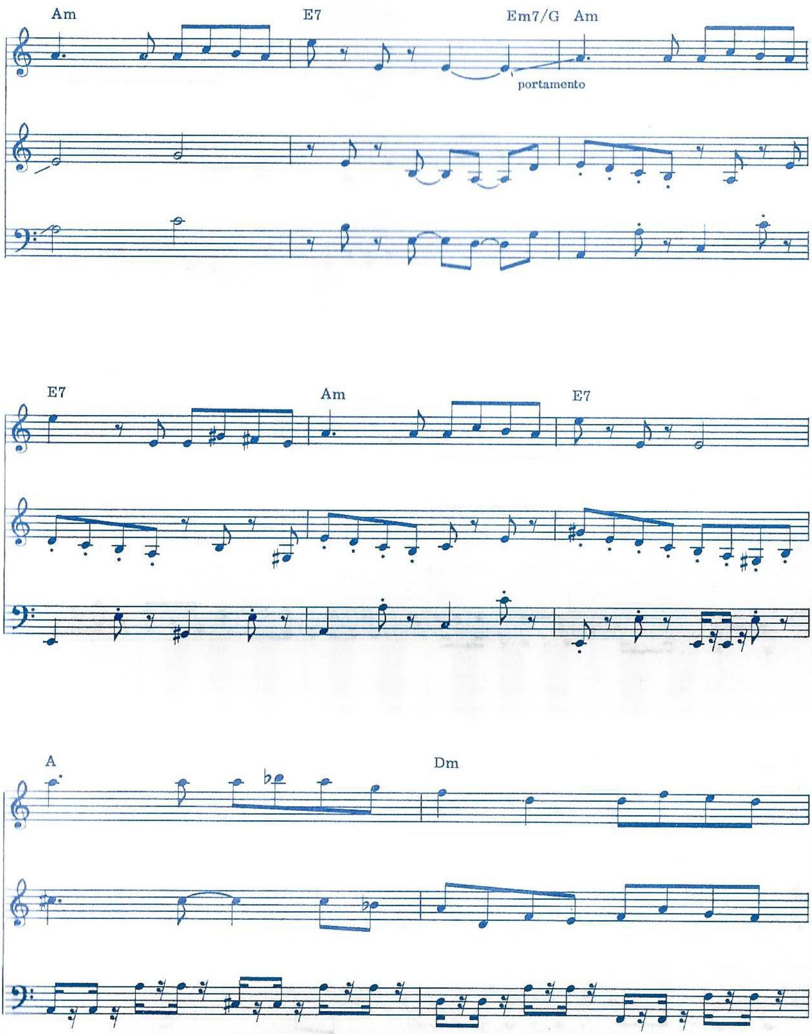
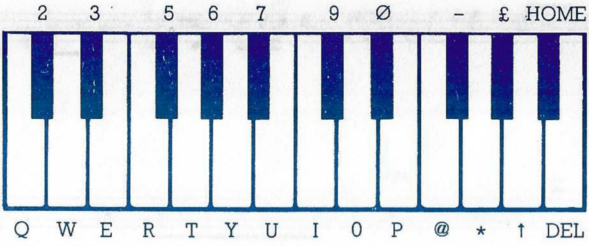

Dm
G7
C
Dm
Am
E7
Am
Am
E7
portamento


Am
E7
Em7/G Am
portamento
E7
Am
E7
A
Dm


G7
C
Dm
Am
E7
Am

The following diagram represents which musical notes correspond to the keys on your computer.

COMPUTER KEY	=	MUSIC NOTE
Q	=	C
2	=	C#
W	=	D
3	=	E♭
E	=	E
R	=	F
5	=	F#
T	=	G
6	=	G#
Y	=	A
7	=	B♭
U	=	B
I	=	C
9	=	C#
0	=	D
Ø	=	E♭
P	=	E
@	=	F
–	=	F#
*	=	G
£	=	G#
↑	=	A
HOME	=	B♭
DEL	=	B



SFX COMPUTER SOFTWARE

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